

STEPHEN TECHNOLOGIES CO.,LTD

SVG500SO 1 FXS + 1 FXO Port

SIP VoIP ATA User Manual

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1 Introduction

This user's manual is for SVG500SO VoIP terminal adapter (TA). This user's manual will explain the IVR instruction, web configuration and command line configuration for the TA. Before using the TA, some setup processes are required to make the TA work properly.

1.1 Hardware Overview

The TA has Networking interfaces, telephone interfaces, LED indication, power connector and Reset button.

- 1.1.1 Two RJ-45 Networking interfaces, these two interfaces support 10/100Mps Fast Ethernet. you can connect one RJ-45 Fast Ethernet port to the ADSL or Switch, and connect the other one to your computer.
- 1.1.2 Two RJ-11 Type analog telephone jack interfaces. Phone port is for user to connect to a phone set and PSTN port is for user to connect to the PSTN Line.
- 1.1.3 LED Indication: There are five LED indicators in the TA to show the Power, network, and Off-Hook indication.
- 1.1.4 Reset Button: Press Reset button for longer than 5 seconds to make the TA restore factory default settings and reboot. If TA works improperly, please make an attempt to Reset and then configure again.

1.2 Software Overview

Network Protocol	Tone
<ul style="list-style-type: none"> • SIP v1 (RFC2543), v2(RFC3261) • IP/TCP/UDP/RTP/RTCP • IP/ICMP/ARP/RARP/SNTP • TFTP Client/DHCP Client/ PPPoE Client • Telnet/HTTP Server • DNS Client • NAT/DHCP Server 	<ul style="list-style-type: none"> • Ring Tone • Ring Back Tone • Dial Tone • Busy Tone • Programming Tone
Codec	Phone Function
<ul style="list-style-type: none"> • G.711: 64k bit/s (PCM) • G.723.1: 6.3k / 5.3k bit/s • G.726: 16k / 24k / 32k / 40k bit/s (ADPCM) • G.729A: 8k bit/s (CS-ACELP) • G.729B: adds VAD & CNG to G.729 	<ul style="list-style-type: none"> • Volume Adjustment • Speed dial key • Phone book • Flash
Voice Quality	IP Assignment
<ul style="list-style-type: none"> • VAD: Voice activity detection • CNG: Comfortable noise generator • LEC: Line echo canceller • Packet Loss Compensation • Adaptive Jitter Buffer 	<ul style="list-style-type: none"> • Static IP • DHCP • PPPoE
Call Function	Security
<ul style="list-style-type: none"> • Call Hold • Call Waiting • Call Forward • Caller ID • 3-way conference 	<ul style="list-style-type: none"> • HTTP 1.1 basic/digest authentication for Web setup • MD5 for SIP authentication (RFC2069/ RFC 2617)
	QoS
	<ul style="list-style-type: none"> • ToS field
	NAT Traversal
	<ul style="list-style-type: none"> • STUN
DTMF Function	Configuration
<ul style="list-style-type: none"> • In-Band DTMF • Out-of Band DTMF • SIP Info 	<ul style="list-style-type: none"> • Web Browser • Console/Telnet • IVR/Keypad
SIP Server	Firmware Upgrade
<ul style="list-style-type: none"> • Registrar Server (three SIP account) • Outbound Proxy 	<ul style="list-style-type: none"> • TFTP • Console • HTTP

2 IVR Interface for The TA

You can use the PSTN phone to configure the TA. Please follow the instruction to configure your terminal adapter.

Group	IVR Action	IVR Menu Choice	Parameter(s)	Notes:
Function	Reboot	#195#	None	After you hear "Option Successful," hang-up. The system will reboot automatically.
Function	Factory Reset	#198#	None	System will automatically Reboot. WARNING: ALL "User-Changeable" NONDEFAULT SETTINGS WILL BE LOST! This will include network and service provider data.
Info	Check IP Address	#120#	None	IVR will announce the current IP address of the TA
Info	Check IP Type	#121#	None	IVR will announce if DHCP is enabled or disabled.
Info	Check the Phone Number	#122#	None	IVR will announce current in use VoIP number
Info	Check Network Mask	#123#	None	IVR will announce the current network mask of the TA.
Info	Check Gateway IP Address	#124#	None	IVR will announce the current gateway IP address of the TA.
Info	Check Primary DNS Server Setting	#125#	None	IVR will announce the current setting in the Primary DNS field.
Info	Check Firmware Version	#128#	None	IVR will announce the version of the firmware running on the TA.
Setting	Set DHCP client	#111#	None	The system will change to DHCP Client type
Setting	Set Static IP Address	#112xxx*xxx*xxx*xxx#	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	DHCP will be disabled and system will change to the Static IP type.
Setting	Set Network Mask	#113xxx*xxx*xxx*xxx#	Enter value using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	Must set Static IP first.
Setting	Set Gateway IP Address	#114xxx*xxx*xxx*xxx#	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	Must set Static IP first.
Setting	Set Primary DNS Server	#115xxx*xxx*xxx*xxx#	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	Must set Static IP first.
Setting	Set Codec	#130+[1-8]#	1:G.711 u-Law, 2: G.711 a-Law, 3: G.723.1, 4: G.729a, 5: G.726 16K, 6: G.726 24K, 7: G.726 32K, 8: G.726 40K,	You can set the codec you want to the first priority.
Setting	Set Handset Gain	#131+[00~15]#	Handset Gain from 0~15	You can set the Handset gain to proper value, default is 6
Setting	Set Handset Volume	#132+[00~12]#	Handset Volume from 0~12	You can set the Handset volume to proper value, default is 10

By default, NAT is on, LAN's ip is 192.168.123.1(dial #120# to check LAN's ip), WAN is DHCP client (dial #126# to check WAN's ip).

3 Setup the TA by Web Browser

The TA provides a built-in web server. You can use Web browser to configure the TA. First please input the IP address in the web browser. In the end of IP address, please add the port number ":9999". Ex:<http://192.168.1.100:9999>. By default, NAT is on, LAN's ip is 192.168.123.1(dial #120# to check LAN's ip), WAN is DHCP client (dial #126# to check WAN's ip). You have to set your PC in the same subnet with TA in order to connect them with each other.

3.1 Login

Please input the username and password into the blank field. The default setting is:

- For Administrator, the username is: root; and the password is: test. If you use this account to login, you can configure all the settings.

2. For normal user, the username is: user; and the password is: test. If you use this account to login, but you can not configure the SIP setting.

Click the "Login" button will move into the TA web based management information page.

If you change the setting in the Web Management interface, please do remember to click the "Submit" button in that page. After you finished the change of the setting, click the "Save" function in the left side, and click the Save Button. When you finished the setting, please click the Reboot function in the left side, and click the Reboot button in that page. After the system restart, all the setting can work properly.

The screenshot shows a 'Login VoIP' form. It has two input fields: 'Username' and 'Password'. Below the fields are two buttons: 'Login' and 'Clear'. At the bottom is a checkbox labeled 'Remember last login'.

3.2 System Information

When you login, you can see the TA's current system information like firmware version, model name etc in this page.

Also you can see the function lists in the left side. You can use mouse to click the function you want to set up.

System Information

This page illustrate the system related information.

Model Name:	VoIP
Firmware Version:	Wed Oct 12 17:08:27 2005.
Codec Version:	Fri Oct 14 17:07:38 2005.

3.3 Phone Book

In Phone Book function, you can setup the Phone Book. The phone book can store up to 140 groups of number.

3.3.1 In phone book, you can add/delete phone number.

3.3.1.1 If you want to add phone number in Phone Book, you need input position, name, URL (ie. Phone number). After input, please click "Add Phone" button.

Phone Book

You could add/delete items in current phone book.

The screenshot shows a 'Phone Book' page. At the top, there is a dropdown menu 'Phone Book Page: page 10'. Below it is a table with columns 'Phone', 'Name', 'URL', and 'Select'. The table contains rows from 90 to 99. At the bottom of the table are three buttons: 'Delete Selected', 'Delete All', and 'Reset'. Below the table is an 'Add New Phone' form with fields for 'Position' (0~139), 'Name', and 'URL'. At the bottom of the form are 'Add Phone' and 'Reset' buttons.

3.3.1.2 To delete a group of number, first select the number, and then click “Delete Selected” button to delete selected number.

3.3.1.3 To delete all number, click “Delete All” button, a dialogue window will show up. Click OK button to delete all numbers.



3.4 Phone Settings

Phone Setting contains Call Forward, SNTP Settings, Volume Settings, DND Settings, Caller ID, Flash Time Settings, Call Waiting Settings and T.38 (Fax) Settings.

3.4.1 Call Forward: you can setup the phone number you want to forward to in this page. There are three type of Forward mode. You can choose between All Forward, Busy Forward, and No Answer Forward by click the corresponding icon.

3.4.1.1 All Forward: All incoming call will forward to the number you choosed. First you have to choose between IP and PSTN. Then you can input the name and the phone number in Name and URL field.

3.4.1.2 Busy Forward: If you are on the phone, the new incoming call will forward to the number you choosed. You can input the name and the phone number in Name and URL field.

3.4.1.3 No Answer Forward: If you can not answer the phone, the incoming call will forward to the number you choosed. You can input the name and the phone number in Name and URL field. Also you have to set No Answer Fwd Time Out for system to start to forward the call to the number you choosed.

When you finished the setting, please click the Submit button.

Forward Setting

You could set the forward number of your phone in this page.

All Forward:	<input checked="" type="radio"/> Off	<input type="radio"/> IP	<input type="radio"/> PSTN
Busy Forward:	<input checked="" type="radio"/> Off	<input type="radio"/> IP	
No Answer Forward:	<input checked="" type="radio"/> Off	<input type="radio"/> IP	<input type="radio"/> PSTN

	Name	URL/Number
All Fwd No.:	<input type="text"/>	<input type="text"/>
Busy Fwd No.:	<input type="text"/>	<input type="text"/>
No Answer Fwd No.:	<input type="text"/>	<input type="text"/>

No Answer Fwd Time Out: 3 (2~8 Ring)

Submit Reset

- 3.4.2 SNTP Settings: you can setup the primary and secondary SNTP Server to get the date/time information. Also you can base on your location to set the Time Zone, and how long is needed to synchronize. When you finished the setting, please click the Submit button.

SNTP Settings

You could set the SNTP servers in this page.

SNTP:	<input checked="" type="radio"/> On <input type="radio"/> Off
Primary Server:	time.windows.com
Secondary Server:	208.184.49.9
Time Zone:	GMT <input type="button" value="+"/> <input type="button" value="08"/> <input type="button" value="00"/> (hh:mm)
Sync. Time:	1 <input type="button" value="0"/> <input type="button" value="0"/> (dd:hh:mm)
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

- 3.4.3 Volume Settings: you can setup the Handset Volume, PSTN-Out Volume, Handset Gain and PSTN-In Gain. When you finished the setting, please click the Submit button.

- 3.4.3.1 Handset Volume is to set the volume for you to hear from the handset.
- 3.4.3.2 PSTN-Out Volume is to set the PSTN volume for you to hear.
- 3.4.3.3 Handset Gain is to set the gain send out to the other side.
- 3.4.3.4 PSTN-In Gain is to set the gain send out to the other side.

Volume Setting

You could set the volume of your phone in this page.

Handset Volume:	9 (0~12)
PSTN-Out Volume:	9 (0~12)
Handset Gain:	14 (0~15)
PSTN-In Gain:	14 (0~15)
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

- 3.4.4 DND Setting function: you can setup the Block Setting to keep the phone silence. You can choose DND Always or DND a period. When you finished the setting, please click the Submit button.

- 3.4.4.1 DND Always: All incoming call will be blocked until disable this feature.
- 3.4.4.2 DND Period: Set a time period and the phone will be blocked during the time period. If the “From” time is large than the “To” time, the “To” time will be regarded as next day’s time.

DND Setting

You could set the do not disturb period of your phone in this page.

DND Always:	<input type="radio"/> On <input checked="" type="radio"/> Off
DND Period:	<input type="radio"/> On <input checked="" type="radio"/> Off
From:	00 : 00 (hh:mm)
To:	00 : 00 (hh:mm)
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

- 3.4.5 Auto Answer function: You can set the Auto Answer function to answer the incoming call by the phone. If the call is come from the IP, then the TA can let user to redial the call to PSTN phone number. If the call is coming from PSTN, then the TA can let user to redial to IP Phone number. After the ring count meet the number you set in Auto Answer Counter, then the auto answer will enable.

Auto Answer

You could enable/disable the auto answer in this page.

Auto Answer:	<input type="radio"/> On <input checked="" type="radio"/> Off
Auto Answer Counter:	<input type="text" value="5"/> (0~8)
PIN Code Enabled:	<input type="radio"/> On <input checked="" type="radio"/> Off
PIN Code:	<input type="text"/>

- 3.4.6 Caller ID function: you can set the device to show Caller ID in your PSTN Phone or IP Phone. There are four selection of Caller ID. You need to base on your environment to set the Caller ID function for FSK or DTMF.

Caller ID Setting

You could enable/disable the caller ID setting in this page.

Caller ID:	<input type="text" value="Don't show caller ID"/>
Single Caller ID:	<input type="radio"/> Yes <input checked="" type="radio"/> No
CID Without Time:	<input type="radio"/> Yes <input checked="" type="radio"/> No

- 3.4.7 Dial Plan: This function provides basic dial number replacement rule.

Dial Plan

You could the set the dial plan in this page.

Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 1:	<input type="text"/> + <input type="text"/>
Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 2:	<input type="text"/> + <input type="text"/>
Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 3:	<input type="text"/> + <input type="text"/>
Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 4:	<input type="text"/> + <input type="text"/>
Auto Dial Time:	<input type="text" value="5"/> (3~9 sec)

Drop prefix: If No, when the number you dialed begins with the number in right blank, the number in left blank will be added automatically before your dialed number. If Yes, when the number you dialed begins with the number in right blank, the number will be replaced with the number in left blank.

Auto Dial Time: the number you dialed will be sent out automatically after the time you set here.

Dial Plan

You could set the dial plan in this page.

Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 1:	002 + 8613+8662
Drop prefix :	<input checked="" type="radio"/> Yes <input type="radio"/> No
Replace rule 2:	006 + 002+003+004+005+007+009
Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 3:	009 + 12
Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 4:	007 + 5xxx+35xx+21xx
Auto Dial Time:	5 (3~9 sec)

Example 1: Drop prefix: No; Replace rule 1: 002, 8613+8662

When your dialed number begins with 8613 or 8662, 002 will be added to the beginning of your dialed number.

Example 2: Drop prefix: Yes, Replace rule 2: 006, 002+003+004+005+007+009

When your dialed number begins with 002 or 003, 004, 005, 007, 009, 002 or 003, 004, 005, 007, 009 will be replaced with 006.

Example 3: Drop prefix: No; Replace rule 4: 007, 5xxx+35xx+21xx

When your dialed number begins with 5 or 35, 21, and is 4 digits, 007 will be added to the beginning of your dialed number.

3.4.8 Flash Time Setting function: Set FXO and FXS flash time.

Flash Time Setting

You could set the flash time in this page.

FXO Flash Time	
Flash Time:	5 x 10MS (9~120)

FXS Flash Time	
Max Flash Time:	60 x 10MS (4~255)

3.4.9 Call Waiting Setting function: You can Enable/Disable the Call Waiting function, When you are talking with someone, there is a new incoming call, you will hear the call waiting tone.

Call Waiting Setting

You could enable/disable the call waiting setting in this page.

Call Waiting: On Off

3.4.10 T.38(Fax) Setting: Enable/disable the fax function and set fax port.

T.38 (FAX) Setting

You could enable/disable the FAX function in this page.

T.38 (FAX):	<input type="radio"/> On <input checked="" type="radio"/> Off
T.38 Port:	61000 (1024~65533)

3.4.11 Hot line setting

Hot line Setting

You could set the hot line in this page.

Use Hot Line : Enable Disable

Hot line number:

If enabled, when you lift handset, the hot line number will be dialed out automatically.

3.4.12 Alarm setting

Alarm Settings

You could set the alarm time in this page.

Alarm: ON OFF

Alarm Time: : (hh:mm)

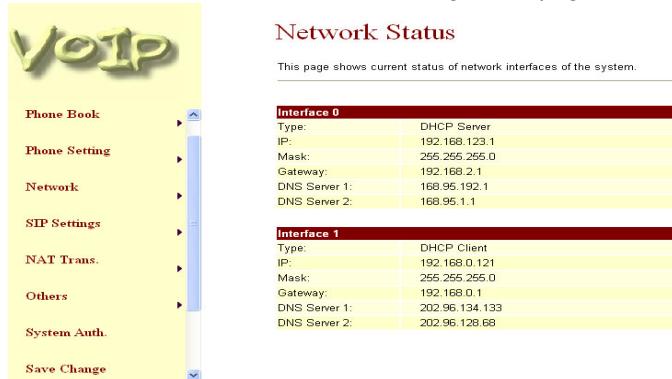
Current time: 2006-10-05 17:47

If On, the phone will alarm at alarm time.

3.5 Network

In Network you can check the Network status, configure the Network Settings, DDNS settings and VLAN settings.

3.5.1 Network Status: You can check the current Network setting in this page.



3.5.2 WAN Settings

- 3.5.2.1 Bridge settings: Enable/disable bridge mode. If you set the Bridge On, then the two Fast Ethernet ports will be transparent
 - 3.5.2.2 WAN Settings: Set network parameters for WAN. WAN can obtain IP address through 3 methods: fixed IP, DHCP client, PPPoE. You may refer to your current network environment to configure the TA properly.
 - 3.5.2.3 The PPPoE Setting: Set PPPoE Username and Password. If you have the PPPoE account from your Service Provider, please input the Username and the Password correctly.
- When you finished the setting, please click the Submit button.

WAN Settings

You could configure the WAN settings in this page.

LAN Mode:	<input checked="" type="radio"/> Bridge <input type="radio"/> NAT
WAN Setting	
IP Type:	<input checked="" type="radio"/> Fixed IP <input type="radio"/> DHCP Client <input type="radio"/> PPPoE
IP:	192.168.1.45
Mask:	255.255.255.0
Gateway:	192.168.1.1
DNS Server1:	202.96.134.133
DNS Server2:	202.96.128.68
MAC:	00167800c52a
PPPoE Setting	
User Name:	<input type="text"/>
Password:	<input type="text"/>
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

- 3.5.3 LAN Settings: Set network parameters for LAN. You may refer to your current network environment to configure the TA properly. When you finished the setting, please click the Submit button.

LAN Settings

You could configure the LAN settings in this page.

LAN Setting	
IP:	<input type="text" value="192.168.123.1"/>
Mask:	<input type="text" value="255.255.255.0"/>
MAC:	<input type="text" value="00167800c52a"/>
DHCP Server	
DHCP Server:	<input type="radio"/> On <input checked="" type="radio"/> Off
Start IP:	<input type="text" value="150"/>
End IP:	<input type="text" value="200"/>
Lease Time:	<input type="text" value="1"/> : <input type="text" value="0"/> (dd:hh)
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

- 3.5.4 DDNS Setting: You can configure the DDNS setting in this page. You need to have the DDNS account and input the informations properly. If you have a DDNS account with a public IP address, others can call you via the DDNS account. But now most of the VoIP applications work with a SIP Proxy Server. When you finished the setting, please click the Submit button.

DDNS Settings

You could set the configuration of DDNS in this page.

DDNS:	<input checked="" type="radio"/> On <input type="radio"/> Off
Host Name:	<input type="text" value="crystalmedia.dyndns.org"/>
User Name:	<input type="text" value="crystalmedia"/>
Password:	<input type="password" value="*****"/>
E-mail Address:	<input type="text"/>
Type:	<input type="text" value="dyndns"/>
Wild Card:	<input type="text" value="on"/>
BACKMX:	<input checked="" type="radio"/> On <input type="radio"/> Off
Off Line:	<input checked="" type="radio"/> On <input type="radio"/> Off
<input type="button" value="Submit"/> <input type="button" value="Reset"/>	

3.5.5 VLAN Settings

VLAN Settings

You could set the VLAN settings in this page.

VLAN Packets:	<input type="radio"/> On <input checked="" type="radio"/> Off
VID:	<input type="text" value="136"/> (2 ~ 4094)
User Priority:	<input type="text" value="0"/> (0 ~ 7)
CFI:	<input type="text" value="0"/> (0 ~ 1)
NAT VLAN Setting	
VLAN Packets:	<input type="radio"/> On <input checked="" type="radio"/> Off
VID1:	<input type="text" value="4"/> (2 ~ 4094), 0->Off
VID2:	<input type="text" value="5"/> (2 ~ 4094), 0->Off
VID3:	<input type="text" value="6"/> (2 ~ 4094), 0->Off
VID4:	<input type="text" value="7"/> (2 ~ 4094), 0->Off

[Submit](#) [Reset](#)

3.5.6 DMZ setting

DMZ Setting

You could configure your demilitarized zone setting in this page.

DMZ:	<input type="radio"/> On <input checked="" type="radio"/> Off
DMZ Host IP:	<input type="text" value="0.0.0.0"/>

[Submit](#) [Reset](#)

3.5.7 Virtual Server

Virtual Server Settings

You could set your virtual servers in this page. The usual port numbers are WEB [TCP 80], FTP (Control) [TCP 21], FTP(Data) [TCP 20], E-mail(POP3) [TCP 110], E-mail(SMTP) [TCP 25], DNS [UDP 53] and Telnet [TCP 23].

Virtual Server Page: [page 1](#) [▼](#)

Num	Enable	Protocol	In Port	Ex Port	Server IP	Select
0	<input type="checkbox"/>					<input type="checkbox"/>
1	<input type="checkbox"/>					<input type="checkbox"/>
2	<input type="checkbox"/>					<input type="checkbox"/>
3	<input type="checkbox"/>					<input type="checkbox"/>
4	<input type="checkbox"/>					<input type="checkbox"/>
5	<input type="checkbox"/>					<input type="checkbox"/>
6	<input type="checkbox"/>					<input type="checkbox"/>
7	<input type="checkbox"/>					<input type="checkbox"/>

[Enable Selected](#) [Delete Selected](#) [Delete All](#) [Reset](#)

Add Virtual Server

Num:	<input type="text" value="0~23"/>		
Server IP:	<input type="text"/>		
Protocol:	<input type="button" value="TCP"/> <input type="button" value="UDP"/>		
Internal Port:	<input type="text"/>	External Port:	<input type="text"/>

[Add Server](#) [Reset](#)

3.6 SIP Settings

In SIP Settings you can setup the Service Domain, Port Settings, Codec Settings, Codec ID Setting, DTMF Setting, RPort Setting and Other Settings. If the VoIP service is provided by ISP, you have to setup the related informations correctly so that you can register to the SIP Proxy Server successfully.

- 3.6.1 In Service Domain Function you need to input the account and the related informations in this page. Please refer to your ISP provider. You can register three SIP accounts in the TA. You can dial the VoIP phone to your friends via first enabled SIP account and receive the phone from these three SIP accounts.

First you need click Active to enable the Service Domain, then you can input the following items:

Display Name: you can input the name you want to display.

User Name: you need to input the User Name got from your ISP.

Register Name: you need to input the Register Name got from your ISP.

Register Password: you need to input the Register Password got from your ISP.

Domain Server: you need to input the Domain Server got from your ISP.

Proxy Server: you need to input the Proxy Server got from your ISP.

Outbound Proxy: you need to input the Outbound Proxy got from your ISP. If your ISP does not provide the information, you can skip this item.

Subscribe for MWI: When set to "On" a Subscribe for Message Waiting Indication will be sent periodically.

Status: You can see the Register Status in the Status item. If the item shows "Registered", then your TA is registered to the ISP, you can make a phone call now.

If you have more than one SIP account, you can follow the steps to register to the other ISP.

When you finished the setting, please click the Submit button.

Service Domain Settings

You could set information of service domains in this page.

Realm 1 (Default)

Active: On Off

Display Name:

User Name:

Register Name:

Register Password:

Domain Server:

Proxy Server:

Outbound Proxy:

Subscribe for MWI: On Off

Status: Not Registered

Realm 2

Active: On Off

Display Name:

User Name:

Register Name:

Register Password:

Domain Server:

Proxy Server:

Outbound Proxy:

Subscribe for MWI: On Off

Status: Not Registered

- 3.6.2 Port Settings: you can setup the SIP and RTP port number in this page. Each ISP provider will have different SIP/RTPport setting, please refer to the ISP to setup the port number correctly. When you finished the setting, please click the Submit button.

Port Settings

You could set the port number in this page.

SIP Port:	<input type="text" value="5060"/> (1024~65535)
RTP Port:	<input type="text" value="60000"/> (1024~65535)

- 3.6.3 Codec Settings: you can setup the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ISP suggestion to setup these items. When you finished the setting, please click the Submit button.

Codec Settings

You could set the codec settings in this page.

Codec Priority
Codec Priority 1: <input type="text" value="G.711 u-law"/>
Codec Priority 2: <input type="text" value="G.711 a-law"/>
Codec Priority 3: <input type="text" value="G.729"/>
Codec Priority 4: <input type="text" value="G.723"/>
Codec Priority 5: <input type="text" value="G.726 - 16"/>
Codec Priority 6: <input type="text" value="G.726 - 24"/>
Codec Priority 7: <input type="text" value="G.726 - 32"/>
Codec Priority 8: <input type="text" value="G.726 - 40"/>

RTP Packet Length
G.711 & G.729: <input type="text" value="20 ms"/>
G.723: <input type="text" value="30 ms"/>

G.723 5.3K
G.723 5.3K: <input type="radio"/> On <input checked="" type="radio"/> Off

Voice VAD
Voice VAD: <input type="radio"/> On <input checked="" type="radio"/> Off

- 3.6.4 Codec ID Setting: Sometimes 2 VoIP devices with different Codec ID will cause some problems. If there are some problems when you are talking with others, you may ask the other one what Codec ID he is using, and change your Codec ID to the same as his. When you finished the setting, please click the Submit button.

Codec ID Setting

You could set the value of Codec ID in this page.

Codec Type	ID	Default Value
G726-16 ID:	23 (95~255)	<input checked="" type="checkbox"/> 23
G726-24 ID:	22 (95~255)	<input checked="" type="checkbox"/> 22
G726-32 ID:	2 (95~255)	<input checked="" type="checkbox"/> 2
G726-40 ID:	21 (95~255)	<input checked="" type="checkbox"/> 21
RFC 2833 ID:	101 (95~255)	<input checked="" type="checkbox"/> 101

- 3.6.5 DTMF Setting: Choose between 2833, Inband DTMF and Send DTMF SIP Info. Please refer to your ISP for correct setting. When you finished the setting, please click the Submit button.

DTMF Setting

You could set the DTMF setting in this page.

2833
 Inband DTMF
 Send DTMF SIP Info

- 3.6.6 RPort Setting: Enable/Disable RPort. To change this setting, please following your ISP's information. When you finished the setting, please click the Submit button.

RPort Setting

You could enable/disable the RPort setting in this page.

RPort: On Off

- 3.6.7 Other Settings: you can setup the Hold by RFC, Voice/SIP QoS and SIP expire time in this page. To change these settings please following your ISP's information. The QoS setting is to set the voice packets' priority. If you set the value higher, the voice packets will get higher priority. But the QoS function still need to cooperate with the others Internet devices. When you finished the setting, please click the Submit button.

Other Settings

You could set other settings in this page.

Hold by RFC:	<input type="radio"/> On <input checked="" type="radio"/> Off
Voice QoS:	40 (0~63)
SIP QoS:	40 (0~63)
SIP Expire Time:	60 (60~86400 sec)

3.7 NAT Trans.

- 3.7.1 STUN Setting: you can setup the STUN Enable/Disable and STUN Server IP address in this page. This function can help your TA working properly behind NAT. To change these settings please following your ISP information. When you finished the setting, please click the Submit button.

STUN Setting

You could set the IP of STUN server in this page.

STUN:	<input type="radio"/> On <input checked="" type="radio"/> Off
STUN Server:	66.7.238.210
STUN Port:	3478 (1024~65535)

3.8 Others

3.8.1 Auto Config: Please set these according to your need.

Auto Configuration Setting

You could enable/disable the auto configuration setting in this page.

Auto Configuration: Off TFTP FTP HTTP

TFTP Server:

HTTP Server:

HTTP Path:

FTP Server:

FTP Username:

FTP Password:

File Path:

3.8.2 FXO & FXS Port: Please set this according to your country's standard.

FXO & FXS Impedance Setting

You could select the FXO & FXS impedance of the analog telephone by different country in this page.

FXO Port:

FXS Port:

3.8.3 MAC Clone Setting: Enable/disable MAC Clone.

MAC Clone Setting

You could enable/disable the MAC clone setting in this page.

MAC Clone: On Off

3.8.4 Tones Settings

Tones Settings

You could configure your tones settings in this page.

	Dial Tone	Ring Back Tone	Busy Tone	Error Tone	Ring Tone	Insert Tone
Cadence On:	<input type="checkbox"/>	<input checked="" type="checkbox"/>				
Hi-Tone Freq.:	440	480	620	620	480	440
Lo-Tone Freq.:	350	440	480	480	440	350
Hi-Tone Gain:	4522	2261	2261	2261	15360	2261
Lo-Tone Gain:	2261	2261	2261	2261	15360	1130
On Time 1:	0	200	50	30	200	30
Off Time 1:	0	400	50	20	400	20
On Time 2:	0	0	0	0	0	30
Off Time 2:	0	0	0	0	0	400
On Time 3:	0	0	0	0	0	0
Off Time 3:	0	0	0	0	0	0

3.8.5 Advanced Settings

Advanced Setting

You could change advanced setting in this page.

ICMP Not Echo:	<input type="radio"/> Yes <input checked="" type="radio"/> No
Send Anonymous CID:	<input type="radio"/> Yes <input checked="" type="radio"/> No
Polarity Reversal:	<input type="radio"/> Yes <input checked="" type="radio"/> No
CPC Delay:	2 <small>(2~5 Seconds)</small>
CPC Duration:	0 <small>x 10MS (0~60)</small>
Send Flash event:	Disabled
SIP Encrypt:	Disabled

3.9 System Auth.

In System Authority you can change your login name and password. When you finished the setting, please click the Submit button.

System Authority

You could change the login username/password in this page.

New username:	<input type="text"/>
New password:	<input type="password"/>
Confirmed password:	<input type="password"/>

3.10 Save Change

In Save Change you can save the changes you have made. If you want to make new setting into effect, You have to click the Save button. After you click the Save button, the TA will automatically restart and the new setting will take effect.

Save Changes

You have to save changes to effect them.

Save Changes:

3.11 Update

In Update you can update the IP Phone's firmware to the latest or restore the factory setting.

3.11.1 In New Firmware function you can update new firmware via Local PC/TFTP in this page. You can upgrade the firmware by the following steps:

- 1) Select the firmware code type, Risc or DSP code.
- 2) Click the “Browse” button in the right side of the File Location, or you can type the correct path and the filename in File Location blank.
- 3) Select the correct file you want to download to the IP Phone then click the Update button.

Update Firmware

You could update the newest firmware.

Method: Local PC TFTP

Local PC	
Code Type:	<input type="button" value="Risc"/>
File Location:	<input type="text"/> 浏览...
TFTP	
TFTP Server:	<input type="text" value="192.168.1.250"/>

3.11.2 Auto Update

Auto Update Settings

You could set auto update settings in this page.

Update via: Off TFTP FTP HTTP

TFTP Server:

HTTP Server: Exp. 60.35.187.30

HTTP File Path: Exp. /download/

FTP Server:

FTP Username:

FTP Password: Exp. 60.35.17.1

FTP File Path: Exp. /file/load

Check new firmware: Power ON Scheduling

Scheduling (Date): 14 (1~30 days)

Scheduling (Time): AM 00:00- 05:59

Automatic Update: Notify only Automatic

Firmware File Prefix: PHONE

- 3.11.3 In Default Setting you can restore the TA to factory settings in this page. You can just click the Restore button, then the TA will restore to default and automatically restart.

Restore Default Settings

You could click the restore button to restore the factory settings.

Restore default settings:

3.12 Reboot

Click the Reboot button, then the TA will reboot automatically.

Reboot System

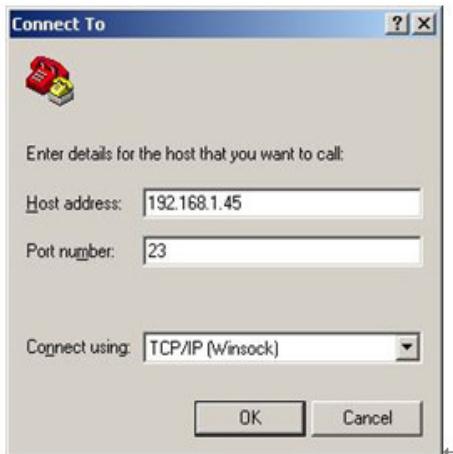
You could press the reboot button to restart the system.

Reboot system:

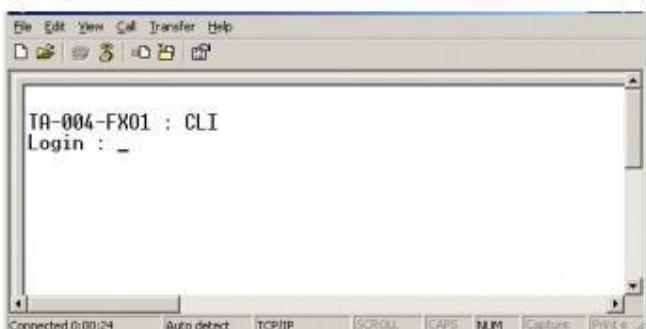
4 Setup the TA by using Console (Hyper Terminal)

4.1 Configuration

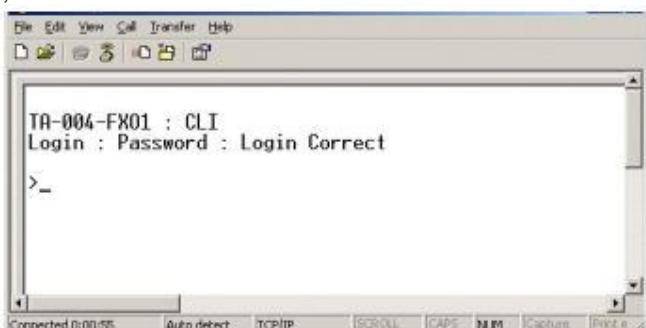
Open the hyper terminal window, input a connection name and click “OK” button. Select TCP/IP(WinSock) in “Connect using” field, input TA’s IP in “Host address” field, “Port number” use the default value 23. Finally click “OK”.



A login window will show up. Input your user name after “Login:” and password after “Password:”. If user name and password are correct, it will prompt “Login Correct”, otherwise will prompt “Login Incorrect”.



After login successfully, the window should be:



4.2 Using CLI command to configure the TA

4.2.1 CLI command list as below:

Itno	Command	Description
1	?	Show CLI Command
2	arp	ARP Configuration
3	ipconfig	Interface Configuration
4	save	Save to flash
5	reboot	Reboot
6	exit	Exit
7	debugmode	Enter Debug Mode

8	update	Update Flash Code/RAM
9	auth	Change User Name and Password
10	nat	NAT Configuration
11	dns	DNS Configuration
12	ping	ping [-IN] [IP-addr host-name]
13	sip	SIP Configuration
14	ddns	DDNS Configuration
15	sntp	SNTP Configuration
16	vlan	VLAN Configuration
17	time	Get System Time
18	mactab	Show MAC Learning Table
19	dump	Read/Write Memory
20	book	Edit phone book
21	reload	Reload Factory Setting
22	watchdog	WatchDog Function
23	phone	Phone Setting
24	weblogo	Change Web's logo
25	dsp	Show dsp type
26	addport	Add Nat Port Mapping
27	cid	Select slic Cid
28	slic	read or write slic registers
29	ver	Firmware Version

4.2.1.1 “?” function is to show CLI command list in the screen.

4.2.1.2 arp function

Itno	Command	Description
1	?	Show 'arp' Option
2	-a	Show ARP Table
3	-d	Delete ARP Table
4	-s	Set Static ARP Table
5	(null)	Show ARP Table

4.2.1.3 ipconfig function

Itno	Command	Description
1	?	Show 'ipconfig' Option
2	-if0	Interface 0
3	-if1	Interface 1
4	-if2	Interface 2
5	-h	Set Host Name
6	-a	Set ARP Cache Expire
7	-r	Restore Current Setting
8	(null)	Show IP Setting

4.2.1.3.1 ipconfig -ifN function → N is 0, 1, 2

Itno	Command	Description
1	?	Show 'ipconfig -ifN' Option
2	-t	Set Host Type
3	-m	Set MAC Address
4	-i	Set IP Address
5	-nm	Set Net Mask
6	-g	Set Gateway
7	-dns0	Set Primary DNS server
8	-dns1	Set Secondary DNS server
9	-dr	Set Default Route
10	-nat	Set NAT
11	on	Enable Interface
12	off	Disable Interface

13	-dhcps	DHCP Server Setting
14	-ddns	Set DDNS
15	-bridge	Set Bridge
16	-dev0	Set Device 0 Setting
17	-dev1	Set Device 1 Setting
18	-dev2	Set Device 2 Setting
19	(null)	Show Interface Setting

4.2.1.4 save function

Itno	Command	Description
1	?	Show 'save' Option
2	-book	Save phone book
3	-sys	Save system setting

4.2.1.5 reboot function is to restart the system.

4.2.1.6 exit function is to exit the CLI.

4.2.1.7 debugmode function is to enter the debugmode.

4.2.1.8 update function

Itno	Command	Description
1	?	Show 'update' Option
2	-os	Update OSImage(IP filename)
3	-dsp	Update DSP Image(IP filename)
4	-all	Update All Image(IP filename)
5	-server	Update Server (IP filename length)
6	-pcm	PCM(IP filename)
	-alaw	alaw (IP filename)
	-ulaw	ulaw (IP filename)
	-g729	g729 (IP filename)
	-g723	g723 (IP filename)
	-g726.16	g726.16 (IP filename)
	-g726.24	g726.24 (IP filename)
	-g726.32	g726.32 (IP filename)
	-g726.40	g726.40 (IP filename)

IP is the TFTP server's IP address, and the filename is the image you want to download into the system.

4.2.1.9 auth function

Itno	Command	Description
1	?	Show 'auth' Option
2	-user	Change User Name.'auth -sys3 -user xxx '
3	-pass	Change Password. 'auth -sys3 -pass xxx xxx'
4	(null)	Show auth's System/PPP Setting

In each item includes

Itno	Command	Description
1	?	Show 'auth' Option
2	-admin	Change Administrator user name/password
3	-sys0	Change System user0 user name/password
4	-sys1	Change System user1 user name/password
5	-sys2	Change System user2 user name/password
6	-sys3	Change System user3 user name/password
7	-sys4	Change System user4 user name/password
8	-norm0	Change Normal user0 user name/password
9	-norm1	Change Normal user1 user name/password
10	-norm2	Change Normal user2 user name/password

11	-norm3	Change Normal user3 user name/password
12	-norm4	Change Normal user4 user name/password
13	-ppp	Change PPP user name/password
14	(null)	Show auth Setting

If you want to change the password, you need to type the password twice in the CLI.

4.2.1.10 nat function

ltno	Command	Description
1	?	Show 'nat' Option
2	-vs	Set 'nat -vs' Option
3	-dmz	Set 'nat -dmz' Option
4	(null)	Show NAT Setting

In DMZ item includes

ltno	Command	Description
1	?	Show 'nat -dmz' Option
2	on	Enable DMZ
3	off	Enable DMZ
4	-ip	Set DMZ IP address
5	(null)	Show DMZ Setting

4.2.1.11 dns function

ltno	Command	Description
1	?	Show 'dns' Option
2	-q	DNS query. dns -q domain-name
3	(null)	Show DNS Table

4.2.1.12 ping function

ltno	Command	Description
1	?	Show 'ping' Option
2	-l	ping [-l N] [IP-addr host-name]
3	(null)	ping [IP-addr host-name]

4.2.1.13 sip function

ltno	Command	Description
1	?	Show 'sip' Option
2	-proxy0	sip -proxy0
3	-proxy1	sip -proxy1
4	-proxy2	sip -proxy2
5	-upnp	sip -upnp on/off/show
6	-exts	sip -exts sip upnp external-port
7	-extr	sip -extr rtp upnp external-port
8	-sipp	sip udp port
9	-rtpp	sip rtp port
10	-stun	sip -stun on/off
11	-rport	sip -rport on/off
12	-sserver	sip -sserver stun-server
13	-out	sip -out outbound-proxy
14	-dump	sip -dump
15	-log	sip -log on/off
16	-drtp	sip -drtp 0/1/2
17	-rtpnc	sip -rtpnc on/off
18	-wanip	sip -wanip
19	-nattype	sip -nattype
20	-hbyrfc	sip -hbyrfc
21	-dereg	sip -dereg
22	-restart	sip -restart
23	-jbt	sip -jitter buffer Threshold

24	(null)	Show SIP Setting
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4.2.1.14 ddns function

Itno	Command	Description
1	?	Show 'ddns' Option
2	-type	Set DDNS Type
3	-host	Set Host Name
4	-wild	Set Wild Card Mode
5	-mx	Set Mail Exchanger
6	-backmx	Set Mail Exchanger Mode
7	-offline	Set Offline Mode
8	-user	Set Login User Name
9	-pass	Set Login Password
10	(null)	Show DDNS Setting

4.2.1.15 sntp function

Itno	Command	Description
1	?	Show 'sntp' Option
2	-on	Enable SNTP Client
3	-off	Disable SNTP Client
4	-ip1	Set SNTP Server1 IP
5	-ip2	Set SNTP Server2 IP
6	-mode	Set SNTP Client Mode
7	-zone	Set GMT Time Zone: [+ -][hour]:[min]
8	-adjust	Set Adjustment Time: [second]
9	(null)	Show SNTP Setting

4.2.1.16 vlan function

Itno	Command	Description
1	?	Show 'vlan' Option
2	-tx	Tx Vlan setting
3	-rx	Rx Vlan setting
4	(null)	Show Vlan Setting

4.2.1.17 time function

Itno	Command	Description
1	?	Show 'Time' Option
2	-t	Modify Time: hour:min:sec
3	-d	Modify date: year:mon:date
4	(null)	Show Data & Time

4.2.1.18 mactab function is to show MAC learning table.

4.2.1.19 dump function

Itno	Command	Description
1	?	Show 'dump' Option
2	-r	dump -r XXXXXXXX
3	-w	dump -w XXXXXXXX XX

4.2.1.20 book function

Itno	Command	Description
1	?	Show 'book' Option
2	-a	Show answer list
3	-c	Show call list
4	-s	speed dial
5	-p	phone book

4.2.1.21 reload function is to Reload Factory Setting, please make sure you want to do the factory reset.

4.2.1.22 watchdog function

Itno	Command	Description
1	?	Show 'WatchDog' Option
2	on	Enable WatchDog
3	off	Disable WatchDog
4	(null)	Show WatchDog Setting

4.2.1.23 phone function

Itno	Command	Description
1	?	Show 'phone' Option
2	-autoanswer	phone auto answer
3	-vol	Volume setting
4	-block	Block Incoming call
5	-ring	Set Melody Ringer
6	-forward	Auto-forward Incall to Phone[0-9] in Book
7	(null)	Show Phone Setting

4.2.1.24 weblogo function

Itno	Command	Description
1	?	Show 'weblogo' Option
2	-on	Vender Logo
3	-off	Crystal media Logo
4	(null)	Show weblogo Setting

4.2.1.25 dsp function is to show dsp code type.

4.2.1.26 addport function is to add Nat Port Mapping

4.2.1.27 cid function

Itno	Command	Description
1	?	Show 'cid' Option
2	-off	Disable Slic Cid signal
3	-1	Tx FSK after 1st Ring
4	-2	Tx FSK before 1st Ring
5	-3	Tx DTMF before 1st Ring
6	-4	Tx FSK with Line reversal before 1st Ring
7	-5	Tx DTMF with Line reversal before 1st Ring
8	-time	FSK cid with time message
9	-single	Single type FSK CID
10	(null)	Show Cid Option

4.2.1.28 slic function

Itno	Command	Description
1	?	Show 'slic' Option
2	-ring	Issue Ring signal
3	-r	read slic addr
4	-w	write slic addr
5	-a	read all slic reg
6	(null)	Show slic register

4.2.1.29 ver function is to show Firmware Version.

5 How to make a phone call

When your TA is configured properly, you can make a phone call to your friend in the same Service provider. Please make sure all the cables are connected properly, like PSTN Line cable, Phone cable, Ethernet cable, power cable.

If you want to make a phone call, you can dial the phone number and press “#” button to start to dial the phone number.

5.1 Dial a PSTN Phone call

Dial “0*” before a phone number. Press “#” button to make the TA send out the dialed number.

5.2 Dial a IP Phone call

Dial IP phone number directly. Press “#” button to make the TA send out the dialed number.

The TA also provides some functions listed as below:

1. Call Waiting: When a new call is coming while you are talking, you can push the Flash button to switch to the new call. You can push the Flash button to switch between the two calls.
2. Call Hold: You can push the Hold key to hold the current call for a while, then push Hold key again to keep talking.
3. 3-way conference: If you want to make a 3-way conference call, you can make a phone call to the first phone number. After the call is established, push the Flash button, dial #512#, and then you can hear the Dial tone, then make a phone call to the second phone number. When the second call is established, press the Flash button again.

6 Get a FWD account

1. From the website www.freeworlddialup.com; you can apply an account to use the VoIP communication. You can follow the instruction to input the information. After you finished, you will receive a mail sent by the FWD mail system, you will get the account information in the mail.
2. When you got the account, you can setup the related information into the TA.
3. You can setup the related information into the TA by web browser. Also you can use Telnet, Console via CLI command to configure the TA. You need to input the Proxy Name, Domain Name, Register Name, and password. The Display Name you can input what you want to let others see.
4. After you registered to the SIP Server, you can try to call your friends who also registered in the same SIP Server. You just need to dial your friend's user name (registered name) and press “#” then you can make a phone call to your friend.

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